Enhancement Speech Matlab Code

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It is your enormously own time to comport yourself reviewing habit. in the course of guides you could enjoy now is **enhancement speech matlab code** below.

Signal Enhancement with Variable Span Linear Filters Jacob Benesty 2018-12-09 This book introduces readers to the novel concept of variable span speech enhancement filters, and demonstrates how it can be used for effective noise reduction in various ways. Further, the book provides the accompanying Matlab code, allowing readers to easily implement the main ideas discussed. Variable span filters combine the ideas of optimal linear filters with those of subspace methods, as they involve the joint diagonalization of the correlation matrices of the desired signal and the noise. The book shows how some well-known filter designs, e.g. the minimum distortion, maximum signal-to-noise ratio, Wiener, and tradeoff filters (including their new generalizations) can be obtained using the variable span filter framework. It then illustrates how the variable span filters can be applied in various contexts, namely in single-channel STFT-based enhancement, in multichannel enhancement in both the time and STFT domains, and, lastly, in time-domain binaural enhancement. In these contexts, the properties of these filters are analyzed in terms of their noise reduction capabilities and desired signal distortion, and the analyses are validated and further explored in simulations.

Audio Watermark Yiqing Lin 2014-09-22 This book illustrates the commonly used and novel approaches of audio watermarking for copyrights protection. The author examines the theoretical and practical step by step guide to the topic of data hiding in audio signal such as music, speech, broadcast. The book covers new techniques developed by the authors are fully explained and MATLAB programs, for audio watermarking and audio quality assessments and also discusses methods for objectively predicting the perceptual quality of the watermarked audio signals. Explains the theoretical basics of the commonly used audio watermarking techniques Discusses the methods used to objectively and subjectively assess the quality of the audio signals Provides a comprehensive well tested MATLAB programs that can be used efficiently to watermark any audio media

Audio and Speech Processing with MATLAB Paul Hill 2018-12-07 Speech and audio processing has undergone a revolution in preceding decades that has accelerated in the last few years generating game-changing technologies such as truly successful speech recognition systems; a goal that had remained out of reach until very recently. This book gives the reader a comprehensive overview of such contemporary speech and audio processing techniques with an emphasis on practical implementations and illustrations using MATLAB code. Core concepts are firstly covered giving an introduction to the physics of audio and vibration together with their representations using complex numbers, Z transforms and frequency analysis transforms such as the FFT. Later chapters give a description of the human auditory system and the fundamentals of psychoacoustics. Insights, results, and analyses given in these chapters are subsequently used as the basis of understanding of the middle section of the book covering: wideband audio compression (MP3 audio etc.), speech recognition and speech coding. The final chapter covers musical synthesis and applications describing methods such as (and giving MATLAB examples of) AM, FM and ring modulation techniques. This chapter gives a final example of the use of time-frequency modification to implement a so-called phase vocoder for time stretching (in MATLAB). Features A comprehensive overview of contemporary speech and audio processing techniques from perceptual and physical acoustic models to a thorough background in relevant digital signal processing techniques together with an exploration of speech and audio applications. A carefully paced progression of complexity of the described methods; building, in many cases, from first principles. Speech and wideband audio coding together with a description of associated standardised codecs (e.g. MP3, AAC and GSM). Speech recognition: Feature extraction (e.g. MFCC features), Hidden Markov Models (HMMs) and deep learning techniques such as Long Short-Time Memory (LSTM) methods. Book and computer-based problems at the end of each chapter. Contains numerous real-world examples backed up by many MATLAB functions and code.

Speech Enhancement Philipos C. Loizou 2013-02-25 With the proliferation of mobile devices and hearing devices, including hearing aids and cochlear implants, there is a growing and pressing need to design algorithms that can improve speech intelligibility without sacrificing quality. Responding to this need, Speech Enhancement: Theory and Practice, Second Edition introduces readers to the basic pr

Practical Image and Video Processing Using MATLAB Oge Marques 2011-08-04 UP-TO-DATE, TECHNICALLY ACCURATE COVERAGE OF ESSENTIAL TOPICS IN IMAGE AND VIDEO PROCESSING This is the first book to combine image and video processing with a practical MATLAB®-oriented approach in order to demonstrate the most important image and video techniques and algorithms. Utilizing minimal math, the contents are presented in a clear, objective manner, emphasizing and encouraging experimentation. The book has been organized into two parts. Part I: Image Processing begins with an overview of the field, then introduces the fundamental concepts, notation, and terminology associated with image representation and basic image processing operations. Next, it discusses MATLAB® and its Image Processing Toolbox with the start of a series of chapters with hands-on activities and step-by-step tutorials. These chapters cover image acquisition and digitization; arithmetic, logic, and geometric operations; point-based, histogram-based, and neighborhood-based image enhancement techniques; the Fourier Transform and relevant frequency-domain image filtering techniques; image restoration; mathematical morphology; edge detection techniques; image segmentation; image compression and coding; and feature extraction and representation. Part II: Video Processing presents the main concepts and terminology associated with analog video signals and systems, as well as digital video formats and standards. It then describes the technically involved problem of standards conversion, discusses motion estimation and compensation techniques, shows how video sequences can be filtered, and concludes with an example of a solution to object detection and tracking in video sequences using MATLAB®. Extra features of this book include: More than 30 MATLAB® tutorials, which consist of step-by-step guides toexploring image and video processing techniques using MATLAB® Chapters supported by figures, examples, illustrative problems, and exercises Useful websites and an extensive list of bibliographical references This accessible text is ideal for upper-level undergraduate and graduate students in digital image and video processing courses, as well as for engineers, researchers, software developers, practitioners, and anyone who wishes to learn about these increasingly popular topics on their own.

Audio Source Separation and Speech Enhancement Emmanuel Vincent 2018-07-24 Learn the technology behind hearing aids, Siri, and Echo Audio source separation and speech enhancement aim to extract one or more source signals of interest from an audio recording involving several sound sources.

These technologies are among the most studied in audio signal processing today and bear a critical role in the success of hearing aids, hands-free phones, voice command and other noise-robust audio analysis systems, and music post-production software. Research on this topic has followed three convergent paths, starting with sensor array processing, computational auditory scene analysis, and machine learning based approaches such as independent component analysis, respectively. This book is the first one to provide a comprehensive overview by presenting the common foundations and the differences between these techniques in a unified setting. Key features: Consolidated perspective on audio source separation and speech enhancement. Both historical perspective and latest advances in the field, e.g. deep neural networks. Diverse disciplines: array processing, machine learning, and statistical signal processing. Covers the most important techniques for both single-channel and multichannel processing. This book provides both introductory and advanced material suitable for people with basic knowledge of signal processing and machine learning. Thanks to its comprehensiveness, it will help students select a promising research track, researchers leverage the acquired cross-domain knowledge to design improved techniques, and engineers and developers choose the right technology for their target application scenario. It will also be useful for practitioners from other fields (e.g., acoustics, multimedia, phonetics, and musicology) willing to exploit audio source separation or speech enhancement as pre-processing tools for their own needs.

New Frontiers in Brain Nawaz Mohamudally 2020-02-26 Brain-Computer Interface (BCI) sounds comparable to plugging a USB cable into a human brain with a laptop and accessing brain information. However, it is not as simple as it sounds. BCI is a multidisciplinary discipline with an exponential progress parallel to and with Artificial Intelligence for the past decades. Initially started with the Electroencephalography (EEG) analysis, BCI offers practical applications for cortical physiology today. Although BCI outcomes are more perceptible in medicine such as cognitive assessment, neurofeedback, and neuroprosthetic implants, it opens up amazing avenues for the business community through machine learning and robotics. Thought-to-text is one example of a hot topic in BCI. So, it is quite predictable to see BCI for individual usage given the current affordability of platforms for less technologically savvy users as well as BCI integrated within office automation productivity tools. The current trend is towards vulgarization for businesses benefits, by extension to the society at large. Thus, the interest in preparing a book on BCI. This book aims to compile and disseminate the latest research findings and best practices on how BCI is expanding the frontiers of knowledge in clinical practices, on the brain itself, and the underlying technologies.

Fundamentals of Signal Enhancement and Array Signal Processing Jacob Benesty 2017-11-13 A comprehensive guide to the theory and practice of signal enhancement and array signal processing, including matlab codes, exercises and instructor and solution manuals Systematically introduces the fundamental principles, theory and applications of signal enhancement and array signal processing in an accessible manner Offers an updated and relevant treatment of array signal processing with rigor and concision Features a companion website that includes presentation files with lecture notes, homework exercises, course projects, solution manuals, instructor manuals, and Matlab codes for the examples in the book

DSP for MATLAB and LabVIEW: LMS adaptive filtering Forester W. Isen 2009 This book is Volume IV of the series DSP for MATLABâ„¢ and LabVIEWâ„¢. Volume IV is an introductory treatment of LMS Adaptive Filtering and applications, and covers cost functions, performance surfaces, coefficient perturbation to estimate the gradient, the LMS algorithm, response of the LMS algorithm to narrow-band signals, and various topologies such as ANC (Active Noise Cancelling) or system modeling, Noise Cancellation, Interference Cancellation, Echo Cancellation (with single- and dual-H topologies), and

Inverse Filtering/Deconvolution. The entire series consists of four volumes that collectively cover basic digital signal processing in a practical and accessible manner, but which nonetheless include all essential foundation mathematics. As the series title implies, the scripts (of which there are more than 200) described in the text and supplied in code form (available via the internet at www.morganclaypool.com/page/isen) will run on both MATLABâ„¢ and LabVIEWâ„¢. The text for all volumes contains many examples, and many useful computational scripts, augmented by demonstration scripts and LabVIEWâ„¢ Virtual Instruments (VIs) that can be run to illustrate various signal processing concepts graphically on the user's computer screen. Volume I consists of four chapters that collectively set forth a brief overview of the field of digital signal processing, useful signals and concepts (including convolution, recursion, difference equations, LTI systems, etc), conversion from the continuous to discrete domain and back (i.e., analog-to-digital and digital-to-analog conversion), aliasing, the Nyquist rate, normalized frequency, sample rate conversion and Mu-law compression, and signal processing principles including correlation, the correlation sequence, the Real DFT, correlation by convolution, matched filtering, simple FIR filters, and simple IIR filters. Chapter 4 of Volume I, in particular, provides an intuitive or "first principle" understanding of how digital filtering and frequency transforms work. Volume II provides detailed coverage of discrete frequency transforms, including a brief overview of common frequency transforms, both discrete and continuous, followed by detailed treatments of the Discrete Time Fourier Transform (DTFT), the z-Transform (including definition and properties, the inverse z-transform, frequency response via z-transform, and alternate filter realization topologies including Direct Form, Direct Form Transposed, Cascade Form, Parallel Form, and Lattice Form), and the Discrete Fourier Transform (DFT) (including Discrete Fourier Series, the DFT-IDFT pair, DFT of common signals, bin width, sampling duration, and sample rate, the FFT, the Goertzel Algorithm, Linear, Periodic, and Circular convolution, DFT Leakage, and computation of the Inverse DFT). Volume III covers digital filter design, including the specific topics of FIR design via windowed-ideal-lowpass filter, FIR highpass, bandpass, and bandstop filter design from windowed-ideal lowpass filters, FIR design using the transition-band-optimized Frequency Sampling technique (implemented by Inverse-DFT or Cosine/Sine Summation Formulas), design of equiripple FIRs of all standard types including Hilbert Transformers and Differentiators via the Remez Exchange Algorithm, design of Butterworth, Chebyshev (Types I and II), and Elliptic analog prototype lowpass filters, conversion of analog lowpass prototype filters to highpass, bandpass, and bandstop filters, and conversion of analog filters to digital filters using the Impulse Invariance and Bilinear Transform techniques. Certain filter topologies specific to FIRs are also discussed, as are two simple FIR types, the Comb and Moving Average filters.

Robust Automatic Speech Recognition Jinyu Li 2015-10-30 Robust Automatic Speech Recognition: A Bridge to Practical Applications establishes a solid foundation for automatic speech recognition that is robust against acoustic environmental distortion. It provides a thorough overview of classical and modern noise-and reverberation robust techniques that have been developed over the past thirty years, with an emphasis on practical methods that have been proven to be successful and which are likely to be further developed for future applications. The strengths and weaknesses of robustness-enhancing speech recognition techniques are carefully analyzed. The book covers noise-robust techniques designed for acoustic models which are based on both Gaussian mixture models and deep neural networks. In addition, a guide to selecting the best methods for practical applications is provided. The reader will: Gain a unified, deep and systematic understanding of the state-of-the-art technologies for robust speech recognition Learn the links and relationship between alternative technologies for robust speech recognition Be able to use the technology analysis and categorization detailed in the book to guide future technology development Be able to develop new noise-robust methods in the current era of deep learning for acoustic modeling in speech recognition The first book that provides a comprehensive review on noise and reverberation robust speech recognition methods in the era of deep neural networks Connects robust speech recognition techniques to machine learning paradigms with rigorous mathematical treatment Provides elegant and structural ways to categorize and analyze noise-robust speech recognition techniques Written by leading researchers who have been actively working on the subject matter in both industrial and academic organizations for many years

Parametric Time-Frequency Domain Spatial Audio Ville Pulkki 2017-10-04 A comprehensive guide that addresses the theory and practice of spatial audio This book provides readers with the principles and best practices in spatial audio signal processing. It describes how sound fields and their perceptual attributes are captured and analyzed within the time-frequency domain, how essential representation parameters are coded, and how such signals are efficiently reproduced for practical applications. The book is split into four parts starting with an overview of the fundamentals. It then goes on to explain the reproduction of spatial sound before offering an examination of signal-dependent spatial filtering. The book finishes with coverage of both current and future applications and the direction that spatial audio research is heading in. Parametric Time-frequency Domain Spatial Audio focuses on applications in entertainment audio, including music, home cinema, and gaming—covering the capturing and reproduction of spatial sound as well as its generation, transduction, representation, transmission, and perception. This book will teach readers the tools needed for such processing, and provides an overview to existing research. It also shows recent up-to-date projects and commercial applications built on top of the systems. Provides an in-depth presentation of the principles, past developments, state-of-the-art methods, and future research directions of spatial audio technologies Includes contributions from leading researchers in the field Offers MATLAB codes with selected chapters An advanced book aimed at readers who are capable of digesting mathematical expressions about digital signal processing and sound field analysis, Parametric Time-frequency Domain Spatial Audio is best suited for researchers in academia and in the audio industry.

Digital Signal Processing for Medical Imaging Using Matlab E.S. Gopi 2012-09-14 This book describes medical imaging systems, such as X-ray, Computed tomography, MRI, etc. from the point of view of digital signal processing. Readers will see techniques applied to medical imaging such as Radon transformation, image reconstruction, image rendering, image enhancement and restoration, and more. This book also outlines the physics behind medical imaging required to understand the techniques being described. The presentation is designed to be accessible to beginners who are doing research in DSP for medical imaging. Matlab programs and illustrations are used wherever possible to reinforce the concepts being discussed.

Digital Speech Processing Using Matlab E. S. Gopi 2013-12-03 Digital Speech Processing Using Matlab deals with digital speech pattern recognition, speech production model, speech feature extraction, and speech compression. The book is written in a manner that is suitable for beginners pursuing basic research in digital speech processing. Matlab illustrations are provided for most topics to enable better understanding of concepts. This book also deals with the basic pattern recognition techniques (illustrated with speech signals using Matlab) such as PCA, LDA, ICA, SVM, HMM, GMM, BPN, and KSOM.

Discrete-Time Speech Signal Processing Thomas F. Quatieri 2008-11-10 Essential principles, practical examples, current applications, and leading-edge research. In this book, Thomas F. Quatieri presents the field's most intensive, up-to-date tutorial and reference on discrete-time speech signal processing. Building on his MIT graduate course, he introduces key principles, essential applications, and state-of-the-art research, and he identifies limitations that point the way to new research opportunities. Quatieri provides an excellent balance of theory and application, beginning with a complete framework for understanding discrete-time speech signal processing. Along the way, he presents important advances never before covered in a speech signal processing text book, including sinusoidal speech processing, advanced time-frequency analysis, and nonlinear aeroacoustic speech production modeling. Coverage includes: Speech production and speech perception: a dual view Crucial distinctions between stochastic and deterministic problems Pole-zero speech models Homomorphic signal processing Short-time Fourier transform analysis/synthesis Filter-bank and wavelet analysis/synthesis Nonlinear measurement and modeling techniques The book's in-depth applications coverage includes speech coding, enhancement, and modification; speaker recognition; noise reduction; signal restoration; dynamic range compression, and more. Principles of Discrete-Time Speech Processing also contains an exceptionally complete series of examples and Matlab exercises, all carefully integrated into the book's coverage of theory and applications.

Speech Enhancement Philipos C. Loizou 2013-02-25 With the proliferation of mobile devices and hearing devices, including hearing aids and cochlear implants, there is a growing and pressing need to design algorithms that can improve speech intelligibility without sacrificing quality. Responding to this need, Speech Enhancement: Theory and Practice, Second Edition introduces readers to the basic problems of speech enhancement and the various algorithms proposed to solve these problems. Updated and expanded, this second edition of the bestselling textbook broadens its scope to include evaluation measures and enhancement algorithms aimed at improving speech intelligibility. Fundamentals, Algorithms, Evaluation, and Future Steps Organized into four parts, the book begins with a review of the fundamentals needed to understand and design better speech enhancement algorithms. The second part describes all the major enhancement algorithms and, because these require an estimate of the noise spectrum, also covers noise estimation algorithms. The third part of the book looks at the measures used to assess the performance, in terms of speech quality and intelligibility, of speech enhancement methods. It also evaluates and compares several of the algorithms. The fourth part presents binary mask algorithms for improving speech intelligibility under ideal conditions. In addition, it suggests steps that can be taken to realize the full potential of these algorithms under realistic conditions. What’s New in This Edition Updates in every chapter A new chapter on objective speech intelligibility measures A new chapter on algorithms for improving speech intelligibility Real-world noise recordings (on accompanying CD) MATLAB® code for the implementation of intelligibility measures (on accompanying CD) MATLAB and C/C++ code for the implementation of algorithms to improve speech intelligibility (on accompanying CD) Valuable Insights from a Pioneer in Speech Enhancement Clear and concise, this book explores how human listeners compensate for acoustic noise in noisy environments. Written by a pioneer in speech enhancement and noise reduction in cochlear implants, it is an essential resource for anyone who wants to implement or incorporate the latest speech enhancement algorithms to improve the quality and intelligibility of speech degraded by noise. Includes a CD with Code and Recordings The accompanying CD provides MATLAB implementations of representative speech enhancement algorithms as well as speech and noise databases for the evaluation of enhancement algorithms.

Fractional Fourier Transform Techniques for Speech Enhancement Prajna Kunche 2020 This book explains speech enhancement in the Fractional Fourier Transform (FRFT) domain and investigates the use of different FRFT algorithms in both single channel and multi-channel enhancement systems, which has proven to be an ideal time frequency analysis tool in many speech signal processing applications. The authors discuss the complexities involved in the highly non- stationary signal processing and the concepts of FRFT for speech enhancement applications. The book explains the fundamentals of FRFT as well as its implementation in speech enhancement. Theories of different FRFT methods are also discussed. The book lets readers understand the new fractional domains to prepare them to develop new algorithms. A comprehensive literature survey regarding the topic is also made available to the reader. Analyzes FRFT techniques in speech enhancement applications; Presents new approaches for speech enhancement using FRFT; Suggests the future directions of research in this emerging area.

Speech Enhancement Shoji Makino 2005-03-17 We live in a noisy world! In all applications (telecommunications, hands-free communications, recording, human-machine interfaces, etc) that require at least one microphone, the signal of interest is usually contaminated by noise and reverberation. As a result, the microphone signal has to be "cleaned" with digital signal processing tools before it is played out, transmitted, or stored. This book is about speech enhancement. Different well-known and state-of-the-art methods for noise reduction, with one or multiple microphones, are discussed. By speech enhancement, we mean not only noise reduction but also dereverberation and separation of independent signals. These topics are also covered in this book. However, the general emphasis is on noise reduction because of the large number of applications that can benefit from this technology. The goal of this book is to provide a strong reference for researchers, engineers, and graduate students who are interested in the problem of signal and speech enhancement. To do so, we invited well-known experts to contribute chapters covering the state of the art in this focused field.

Applied Speech and Audio Processing Ian McLoughlin 2009-02-19 This hands-on, one-stop resource describes the key techniques of speech and audio processing illustrated with extensive MATLAB examples.

Audio Signal Processing and Coding Andreas Spanias 2006-09-11 An in-depth treatment of algorithms and standards for perceptual coding of high-fidelity audio, this self-contained reference surveys and addresses all aspects of the field. Coverage includes signal processing and perceptual (psychoacoustic) fundamentals, details on relevant research and signal models, details on standardization and applications, and details on performance measures and perceptual measurement systems. It includes a comprehensive bibliography with over 600 references, computer exercises, and MATLAB-based projects for use in EE multimedia, computer science, and DSP courses. An ftp site containing supplementary material such as wave files, MATLAB programs and workspaces for the students to solve some of the numerical problems and computer exercises in the book can be found at ftp://ftp.wiley.com/public/sci_tech_med/audio_signal

Deep Learning Applications M. Arif Wani 2020-02-29 This book presents a compilation of selected papers from the 17th IEEE International Conference on Machine Learning and Applications (IEEE ICMLA 2018), focusing on use of deep learning technology in application like game playing, medical applications, video analytics, regression/classification, object detection/recognition and robotic control in industrial environments. It highlights novel ways of using deep neural networks to solve real-world problems, and also offers insights into deep learning architectures and algorithms, making it an essential reference guide for academic researchers, professionals, software engineers in industry, and innovative product developers.

Single Channel Phase-Aware Signal Processing in Speech Communication Pejman Mowlaee 2016-10-18 An overview on the challenging new topic of phase-aware signal processing Speech communication technology is a key factor in human-machine interaction, digital hearing aids, mobile telephony, and automatic speech/speaker recognition. With the proliferation of these applications, there is a growing requirement for advanced methodologies that can push the limits of the conventional solutions relying on processing the signal magnitude spectrum. Single-Channel Phase-Aware Signal Processing in Speech

Communication provides a comprehensive guide to phase signal processing and reviews the history of phase importance in the literature, basic problems in phase processing, fundamentals of phase estimation together with several applications to demonstrate the usefulness of phase processing. Key features: Analysis of recent advances demonstrating the positive impact of phase-based processing in pushing the limits of conventional methods. Offers unique coverage of the historical context, fundamentals of phase processing and provides several examples in speech communication. Provides a detailed review of many references and discusses the existing signal processing techniques required to deal with phase information in different applications involved with speech. The book supplies various examples and MATLAB® implementations delivered within the PhaseLab toolbox. Single-Channel Phase-Aware Signal Processing in Speech Communication is a valuable single-source for students, non-expert DSP engineers, academics and graduate students.

Digital Signal and Image Processing Using MATLAB Maurice Charbit 2010-01-05 This title provides the most important theoretical aspects of Image and Signal Processing (ISP) for both deterministic and random signals. The theory is supported by exercises and computer simulations relating to real applications. More than 200 programs and functions are provided in the MATLAB® language, with useful comments and guidance, to enable numerical experiments to be carried out, thus allowing readers to develop a deeper understanding of both the theoretical and practical aspects of this subject.

Real-Time Digital Signal Processing Sen M. Kuo 2006-05-01 Real-time Digital Signal Processing: Implementations and Applications has been completely updated and revised for the 2nd edition and remains the only book on DSP to provide an overview of DSP theory and programming with hands-on experiments using MATLAB, C and the newest fixed-point processors from Texas Instruments (TI).

Digital Signal and Image Processing Tamal Bose 2004 Introducing the first text to integrate the topics of digital signal processing (DSP), digital image processing (DIP), and adaptive signal processing (ASP)! Digital Signal and Image Processing helps students develop a well-rounded understanding of these key areas by focusing on fundamental concepts, mathematical foundations, and advanced algorithms. The presentation is mathematically thorough with clear explanations, numerous examples, illustrations, and applications. In addition to problems, MATLAB-based computer projects are assigned at the end of each chapter, making this book ideal for laboratory-based courses.

Speech Dereverberation Patrick A. Naylor 2010-07-27 Speech Dereverberation gathers together an overview, a mathematical formulation of the problem and the state-of-the-art solutions for dereverberation. Speech Dereverberation presents current approaches to the problem of reverberation. It provides a review of topics in room acoustics and also describes performance measures for dereverberation. The algorithms are then explained with mathematical analysis and examples that enable the reader to see the strengths and weaknesses of the various techniques, as well as giving an understanding of the questions still to be addressed. Techniques rooted in speech enhancement are included, in addition to a treatment of multichannel blind acoustic system identification and inversion. The TRINICON framework is shown in the context of dereverberation to be a generalization of the signal processing for a range of analysis and enhancement techniques. Speech Dereverberation is suitable for students at masters and doctoral level, as well as established researchers.

Mathematical Modeling and Signal Processing in Speech and Hearing Sciences Jack Xin 2014-04-14 The aim of the book is to give an accessible introduction of mathematical models and signal processing methods in speech and hearing sciences for senior undergraduate and beginning graduate students with basic knowledge of linear algebra, differential equations, numerical analysis, and probability. Speech and hearing sciences are fundamental to numerous technological advances of the digital world in the past decade, from music compression in MP3 to digital hearing aids, from network based voice enabled services to speech interaction with mobile phones. Mathematics and computation are intimately related to these leaps and bounds. On the other hand, speech and hearing are strongly interdisciplinary areas where dissimilar scientific and engineering publications and approaches often coexist and make it difficult for newcomers to enter.

Digital Signal Processing Using MATLAB for Students and Researchers John W. Leis 2011-10-14 Quickly Engages in Applying Algorithmic Techniques to Solve Practical Signal Processing Problems With its active, hands-on learning approach, this text enables readers to master the underlying principles of digital signal processing and its many applications in industries such as digital television, mobile and broadband communications, and medical/scientific devices. Carefully developed MATLAB® examples throughout the text illustrate the mathematical concepts and use of digital signal processing algorithms. Readers will develop a deeper understanding of how to apply the algorithms by manipulating the codes in the examples to see their effect. Moreover, plenty of exercises help to put knowledge into practice solving real-world signal processing challenges. Following an introductory chapter, the text explores: Sampled signals and digital processing Random signals Representing signals and systems Temporal and spatial signal processing Frequency analysis of signals Discrete-time filters and recursive filters Each chapter begins with chapter objectives and an introduction. A summary at the end of each chapter ensures that one has mastered all the key concepts and techniques before progressing in the text. Lastly, appendices listing selected web resources, research papers, and related textbooks enable the investigation of individual topics in greater depth. Upon completion of this text, readers will understand how to apply key algorithmic techniques to address practical signal processing problems as well as develop their own signal processing algorithms. Moreover, the text provides a solid foundation for evaluating and applying new digital processing signal techniques as they are developed.

Speech and Audio Processing Ian Vince McLoughlin 2016-07-21 With this comprehensive and accessible introduction to the field, you will gain all the skills and knowledge needed to work with current and future audio, speech, and hearing processing technologies. Topics covered include mobile telephony, human-computer interfacing through speech, medical applications of speech and hearing technology, electronic music, audio compression and reproduction, big data audio systems and the analysis of sounds in the environment. All of this is supported by numerous practical illustrations, exercises, and hands-on MATLAB® examples on topics as diverse as psychoacoustics (including some auditory illusions), voice changers, speech compression, signal analysis and visualisation, stereo processing, low-frequency ultrasonic scanning, and machine learning techniques for big data. With its pragmatic and application driven focus, and concise explanations, this is an essential resource for anyone who wants to rapidly gain a practical understanding of speech and audio processing and technology.

Speech Enhancement Philipos C. Loizou 2017-03-31 With the proliferation of mobile devices and hearing devices, including hearing aids and cochlear implants, there is a growing and pressing need to design algorithms that can improve speech intelligibility without sacrificing quality. Responding to this need, Speech Enhancement: Theory and Practice, Second Edition introduces readers to the basic problems of speech enhancement and the various algorithms proposed to solve these problems. Updated and expanded, this second edition of the bestselling textbook broadens its scope to include evaluation measures and enhancement algorithms aimed at improving speech intelligibility. Fundamentals, Algorithms, Evaluation, and Future Steps Organized into four parts, the book begins with a review of the fundamentals needed to understand and design better speech enhancement algorithms. The second part describes all the major enhancement algorithms and, because these require an estimate of the noise spectrum, also covers noise estimation algorithms. The third part of the book looks at the measures used to assess the performance, in terms of speech quality and intelligibility, of speech enhancement methods. It also evaluates and compares several of the algorithms. The fourth part presents binary mask algorithms for improving speech intelligibility under ideal conditions. In addition, it suggests steps that can be taken to realize the full potential of these algorithms under realistic conditions. What�s New in This Edition Updates in every chapter A new chapter on objective speech intelligibility measures A new chapter on algorithms for improving speech intelligibility Real-world noise recordings (on accompanying CD) MATLAB� code for the implementation of intelligibility measures (on accompanying CD) MATLAB and C/C++ code for the implementation of algorithms to improve speech intelligibility (on accompanying CD) Valuable Insights from a Pioneer in Speech Enhancement Clear and concise, this book explores how human listeners compensate for acoustic noise in noisy environments. Written by a pioneer in speech enhancement and noise reduction in cochlear implants, it is an essential resource for anyone who wants to implement or incorporate the latest speech enhancement algorithms to improve the quality and intelligibility of speech degraded by noise. Includes a CD with Code and Recordings The accompanying CD provides MATLAB implementations of representative speech enhancement algorithms as well as speech and noise databases for the evaluation of enhancement algorithms.

Sound Capture and Processing Ivan Jelev Tashev 2009-07-01 Provides state-of-the-art algorithms for sound capture, processing and enhancement Sound Capture and Processing: Practical Approaches covers the digital signal processing algorithms and devices for capturing sounds, mostly human speech. It explores the devices and technologies used to capture, enhance and process sound for the needs of communication and speech recognition in modern computers and communication devices. This book gives a comprehensive introduction to basic acoustics and microphones, with coverage of algorithms for noise reduction, acoustic echo cancellation, dereverberation and microphone arrays; charting the progress of such technologies from their evolution to present day standard. Sound Capture and Processing: Practical Approaches Brings together the state-of-the-art algorithms for sound capture, processing and enhancement in one easily accessible volume Provides invaluable implementation techniques required to process algorithms for real life applications and devices Covers a number of advanced sound processing techniques, such as multichannel acoustic echo cancellation, dereverberation and source separation Generously illustrated with figures and charts to demonstrate how sound capture and audio processing systems work An accompanying website containing Matlab code to illustrate the algorithms This invaluable guide will provide audio, R&D and software engineers in the industry of building systems or computer peripherals for speech enhancement with a comprehensive overview of the technologies, devices and algorithms required for modern computers and communication devices. Graduate students studying electrical engineering and computer science, and researchers in multimedia, cell-phones, interactive systems and acousticians will also benefit from this book.

Speech Enhancement Jacob Benesty 2006-03-30 A strong reference on the problem of signal and speech enhancement, describing the newest developments in this exciting field. The general emphasis is on noise reduction, because of the large number of applications that can benefit from this technology.

Blind Speech Separation Shoji Makino 2007-09-07 This is the world’s first edited book on independent component analysis (ICA)-based blind source separation (BSS) of convolutive mixtures of speech. This book brings together a small number of leading researchers to provide tutorial-like and in-depth treatment on major ICA-based BSS topics, with the objective of becoming the definitive source for current, comprehensive, authoritative, and yet accessible treatment.

DFT-Domain Based Single-Microphone Noise Reduction for Speech Enhancement Richard C. Hendriks 2013-01-01 As speech processing devices like mobile phones, voice controlled devices, and hearing aids have increased in popularity, people expect them to work anywhere and at any time without user intervention. However, the presence of acoustical disturbances limits the use of these applications, degrades their performance, or causes the user difficulties in understanding the conversation or appreciating the device. A common way to reduce the effects of such disturbances is through the use of single-microphone noise reduction algorithms for speech enhancement. The field of single-microphone noise reduction for speech enhancement comprises a history of more than 30 years of research. In this survey, we wish to demonstrate the significant advances that have been made during the last decade in the field of discrete Fourier transform domain-based single-channel noise reduction for speech enhancement.Furthermore, our goal is to provide a concise description of a state-of-the-art speech enhancement system, and demonstrate the relative importance of the various building blocks of such a system. This allows the non-expert DSP practitioner to judge the relevance of each building block and to implement a close-to-optimal enhancement system for the particular application at hand. Table of Contents: Introduction / Single Channel Speech Enhancement: General Principles / DFT-Based Speech Enhancement Methods: Signal Model and Notation / Speech DFT Estimators / Speech Presence Probability Estimation / Noise PSD Estimation / Speech PSD Estimation / Performance Evaluation Methods / Simulation Experiments with Single-Channel Enhancement Systems / Future Directions

Subband Adaptive Filtering Kong-Aik Lee 2009-07-06 Subband adaptive filtering is rapidly becoming one of the most effective techniques for reducing computational complexity and improving the convergence rate of algorithms in adaptive signal processing applications. This book provides an introductory, yet extensive guide on the theory of various subband adaptive filtering techniques. For beginners, the authors discuss the basic principles that underlie the design and implementation of subband adaptive filters. For advanced readers, a comprehensive coverage of recent developments, such as multiband tap–weight adaptation, delayless architectures, and filter–bank design methods for reducing band–edge effects are included. Several analysis techniques and complexity evaluation are also introduced in this book to provide better understanding of subband adaptive filtering. This book bridges the gaps between the mixed–domain natures of subband adaptive filtering techniques and provides enough depth to the material augmented by many MATLAB® functions and examples. Key Features: Acts as a timely introduction for researchers, graduate students and engineers who want to design and deploy subband adaptive filters in their research and applications. Bridges the gaps between two distinct domains: adaptive filter theory and multirate signal processing. Uses a practical approach through MATLAB®-based source programs on the accompanying CD. Includes more than 100 M-files, allowing readers to modify the code for different algorithms and applications and to gain more insight into the theory and concepts of subband adaptive filters. Subband Adaptive Filtering is aimed primarily at practicing engineers, as well as senior undergraduate and graduate students. It will also be of interest to researchers, technical managers, and computer scientists.

Noise Reduction in Speech Applications Gillian M. Davis 2018-10-03 Noise and distortion that degrade the quality of speech signals can come from any number of sources. The technology and techniques for dealing with noise are almost as numerous, but it is only recently, with the development of inexpensive digital signal processing hardware, that the implementation of the technology has become practical. Noise Reduction in Speech Applications provides a comprehensive introduction to modern techniques for removing or reducing background noise from a range of speech-related applications. Self-contained, it starts with a tutorial-style chapter of background material, then focuses on system aspects, digital algorithms, and implementation. The final section explores a variety of applications and demonstrates to potential users of the technology the results possible with the noise reduction techniques presented. The book offers chapters contributed by international experts, a practical, systems approach, and numerous references. For electrical, acoustics, signal processing, communications, and bioengineers, Noise Reduction in Speech Applications is a valuable resource that shows you how to decide whether noise reduction will solve problems in your own systems and how to make the best use of the technologies available.

An Introduction to Kalman Filtering with MATLAB Examples Narayan Kovvali 2022-06-01 The Kalman filter is the Bayesian optimum solution to the problem of sequentially estimating the states of a dynamical system in which the state evolution and measurement processes are both linear and Gaussian. Given the ubiquity of such systems, the Kalman filter finds use in a variety of applications, e.g., target tracking, guidance and navigation, and communications systems. The purpose of this book is to present a brief introduction to Kalman filtering. The theoretical framework of the Kalman filter is first presented, followed by examples showing its use in practical applications. Extensions of the method to nonlinear problems and distributed applications are discussed. A software implementation of the algorithm in the MATLAB programming language is provided, as well as MATLAB code for several example applications discussed in the manuscript.

Advanced Signal Processing and Digital Noise Reduction Saeed V. Vaseghi 1996-07-25 Noise cancellation is particularly important in the new mobile communications field, with respect to background noise and acoustic interference in moving vehicles. This comprehensive text develops a coherent and structured presentation of a broad range of the theory and application of statistical signal processing, with emphasis on digital noise reduction algorithms. Other applications covered are spectral estimation, channel equalisation, speech coding over noisy channels, speech recognition in adverse environments, active noise control, echo cancellation, restoration of lost filters, and adaptive notch filters.

Digital Signal Processing Using MATLAB Vinay K. Ingle 2007 This supplement to any standard DSP text is one of the first books to successfully integrate the use of MATLAB® in the study of DSP concepts. In this book, MATLAB® is used as a computing tool to explore traditional DSP topics, and solve problems to gain insight. This greatly expands the range and complexity of problems that students can effectively study in the course. Since DSP applications are primarily algorithms implemented on a DSP processor or software, a fair amount of programming is required. Using interactive software such as MATLAB® makes it possible to place more emphasis on learning new and difficult concepts than on programming algorithms. Interesting practical examples are discussed and useful problems are explored. This updated second edition includes new homework problems and revises the scripts in the book, available functions, and m-files to MATLAB® V7.

Signal Enhancement with Variable Span Linear Filters Jacob Benesty 2016-02-05 This book introduces readers to the novel concept of variable span speech enhancement filters, and demonstrates how it can be used for effective noise reduction in various ways. Further, the book provides the accompanying Matlab code, allowing readers to easily implement the main ideas discussed. Variable span filters combine the ideas of optimal linear filters with those of subspace methods, as they involve the joint diagonalization of the correlation matrices of the desired signal and the noise. The book shows how some well-known filter designs, e.g. the minimum distortion, maximum signal-to-noise ratio, Wiener, and tradeoff filters (including their new generalizations) can be obtained using the variable span filter framework. It then illustrates how the variable span filters can be applied in various contexts, namely in single-channel STFT-based enhancement, in multichannel enhancement in both the time and STFT domains, and, lastly, in time-domain binaural enhancement. In these contexts, the properties of these filters are analyzed in terms of their noise reduction capabilities and desired signal distortion, and the analyses are validated and further explored in simulations.

Digital Signal Processing Lizhe Tan 2013-01-21 Digital Signal Processing, Second Edition enables electrical engineers and technicians in the fields of biomedical, computer, and electronics engineering to master the essential fundamentals of DSP principles and practice. Many instructive worked examples are used to illustrate the material, and the use of mathematics is minimized for easier grasp of concepts. As such, this title is also useful to undergraduates in electrical engineering, and as a reference for science students and practicing engineers. The book goes beyond DSP theory, to show implementation of algorithms in hardware and software. Additional topics covered include adaptive filtering with noise reduction and echo cancellations, speech compression, signal sampling, digital filter realizations, filter design, multimedia applications, over-sampling, etc. More advanced topics are also covered, such as adaptive filters, speech compression such as PCM, u-law, ADPCM, and multi-rate DSP and over-sampling ADC. New to this edition: MATLAB projects dealing with practical applications added throughout the book New chapter (chapter 13) covering sub-band coding and wavelet transforms, methods that have become popular in the DSP field New applications included in many chapters, including applications of DFT to seismic signals, electrocardiography data, and vibration signals All real-time C programs revised for the TMS320C6713 DSK Covers DSP principles with emphasis on communications and control applications Chapter objectives, worked examples, and end-of-chapter exercises aid the reader in grasping key concepts and solving related problems Website with MATLAB programs for simulation and C programs for real-time DSP